X-207-71-9
PREPRINT

NASA TM X- 65426

FINAL TECHNICAL REPORT ON PHONOCARDIOGRAPHIC PREPROCESSOR

JUNE 22 – AUGUST 28, 1970

JANUARY 1971

GSFC

GODDARD SPACE FLIGHT CENTER
GREENBELT, MARYLAND

X-207-71-9

FINAL TECHNICAL REPORT
ON
PHONOCARDIOGRAPHIC PREPROCESSOR

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January 1971

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PREFACE

The technical results presented here represent a ten week effort by its authors during the Summer Institute for Biomedical Research sponsored by the Technology Utilization Office at the Goddard Space Flight Center. Their challenge was to apply NASA developed technology toward the solution of this particular problem and to demonstrate its usefulness to other problems in medical diagnostic monitoring instrumentation.

This report has been published and made available for general use so that others in both the technical and medical communities might benefit from the work of these individuals.

Wayne T. Chen

Wayne T. Chen, Coordinator
Summer Institute for Biomedical Research
Technology Utilization Office

June 29, 1970

PROPOSED PROJECT

After reviewing the previous work, either design a new circuit to produce heart sound envelopes, or improve the existing one. Design requirements are to provide frequency information to some extent, calibrate the signal for true intensity and produce a portable prototype, preferably of very small size. The circuit which meets these requirements should thereby make it feasible to identify either by physician or computer the interpretations listed on page of this notebook.

Routine clinical evaluation of the phonocardiographic amplitude/time signal is subject to interpretation of the complex waveform which carries a full continuum of frequency information. The true intensity and frequency relations are very difficult to perceive in this situation. A circuit which would produce an intensity (power) signal from the amplitude (voltage) signal would provide information better related to what the physician heard, i.e., loudness of sounds and their temporal relations. Frequency related information, analogous to pitch of audible heart sounds would need to be retained to some extent, but not to the equivalent of spectral analysis methods.

Discussion with Dave Winer from George Washington University on the sound envelope project. Topic: Preprocessor for Health Sounds Output of Preprocessor: instantaneous value of intensity with respect to time.

The output of the preprocessor must have the following characteristics:

1. Gives the physician a picture of what he hears

2. Gives the computer a waveform that can be easily stored and analyzed.

The output of the preprocessor can be shifted in time from the real sounds—although it is desirable not to have a time shift.

In the final design the preprocessor should be small enough to connect to the microphone or to a recording unit.

July 1, 1970

Notes from: Winer, et al.; Heart Sound Analysis: A Three Dimensional Approach; The American Journal of Cardiology, Vol. 16, October, 1965

"It is difficult to relate audible intensity of complex sounds to instantaneous values of amplitude as portrayed in the oscillographic form."

"Visible fluctuation in amplitude may not in many instances correspond to audible sensations because the ear responds to root-mean-square pressure, which means that sound must persist for some duration in order to be perceptible. Some transients visible on the oscillograph, therefore, may be of insufficient duration for audible recognition."

Notes from: Winer, David; Notebook: Phonocardiogram Project, December 1, 1964.

"A simplified waveform can be obtained which will include the same information which is presently used and needed. But it should have these advantages:

1. easier to program

2. easier to interpret visually

3. slower sampling rate

    a. compressed feed time to computer

    b. data phone transmission"

<u>Method</u>

"Before the PCG signal is digitized it can be full wave rectified. The next stage in the circuit would be introduction of a suitable time constant to prevent return to baseline (this constant can be adjusted for the desired resolution for timing of split sounds, etc.) This is essentially the technique described by:" Rushmer, et al.; "Sonvelographic Recording of Murmurs During Acute Myocarditis:" American Heart Journal, 48:835, 1954.

"The result is still an <u>amplitude</u> curve, which does not divulge intensity. If one wished for an <u>intensity/time</u> curve, this could be accomplished by a circuit similar to that in a vacuum tube voltmeter which has output calibrated in decibels. This output signal could be put on tape at the data acquisition system or could be generated at the computer center immediately prior to digitizing."

Note added: (February 19, 1965)

"Speech data are carried largely by the varying shape of the power density spectrum and not—as many wrongly believe—in the sound pressure vs time plot seen on an oscilloscope."

S. E. Gerber and E. J. Strausman ("speech scientists," Communication Division Hughes A/C Co., 5440 W. Century, Los Angeles, California) in Digital coding for wire communication, Space/Aeronautics v. 40, No. 5, October, 1963, pp. 95-96.

(February 25, 1965) "A baseline was determined by finding the MODE of all values of amplitude. (The mode is that value which appears most frequently.) We have found that this makes an excellent baseline - Don Sherman's contribution."

(July 29, 1966) "If an analog network were designed to create this type of output (visual records of the way heart-sounds sound to the ear) from the microphone amplitude signal, it could be handled much the same as other low-frequency signals, such as ECG, PTG, EEG, etc. We could expect to find the following benefits:

- 500 digitizing rate vs 3000
- FM channel recording
- Easy computer pattern recognition program
- Telemetry over standard circuits
- Lower pulse code modulation rate if NASA would like to send heart-sounds from space
- Less computer care"

(November 31, 1968) Referring to Mark Wilber's circuit:

"This technique can be used to 'preprocess' heart sounds before recording, allowing all data acquisition, telemetry, preprocessing and processing the computer using the standard MSDL ECG hardware."

(December 4, 1968) Comment about computer analysis:

"Note that this is a digital computer approach to the intensity curves of pp. 6, 7, and 8 of this book. The analog preprocessor saves MUCH computer time."

Perry et al.; Computer Analysis of the Phonocardiogram; reprinted from Engineering in the Practice of Medicine; The Williams and Wilkins Company, 1967.

"Rectification and smoothing techniques may aid in more precise identification of the heart sounds in the presence of artifacts or murmurs."

July 2, 1970

Siebert; from notes given in course titled "Signals and Systems" given at MIT in Spring, 1969. Notes were written in 1967.

Square-law and other non-linear-resistive devices

An ideal square-law device is described by the formula

$$Y(+) = A\,x^2\,(+) \tag{2}$$

(where A is a constant) and is often used to represent approximately the input-output characteristics of full-wave rectifier circuits such as shown in Figure 1.

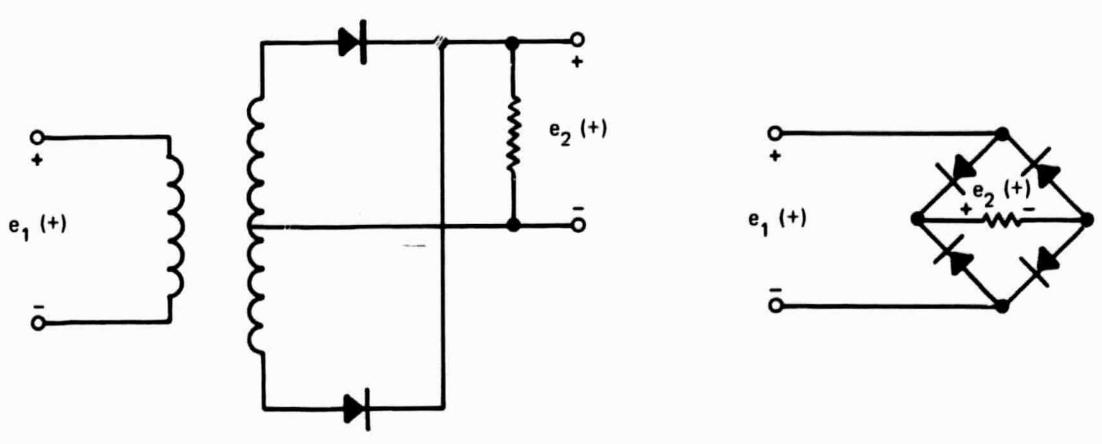


Figure 1. Two Examples of Full-Wave Rectifier Circuits

Other common devices which might be represented by (2) are the temperature of the heater element in a thermocouple-type ammeter as a function of the current, and the output of photomultiplier illuminated by a laser. The actual input-output characteristics of such devices might more accurately be described as in Figure 2. The extent to which such a graph will be adequately represented by (2) depends, of course, on the particular diodes employed, on the amplitude of the signals and on the precision required in the representation.

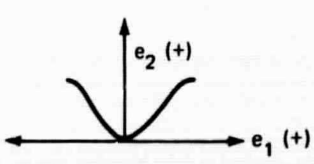
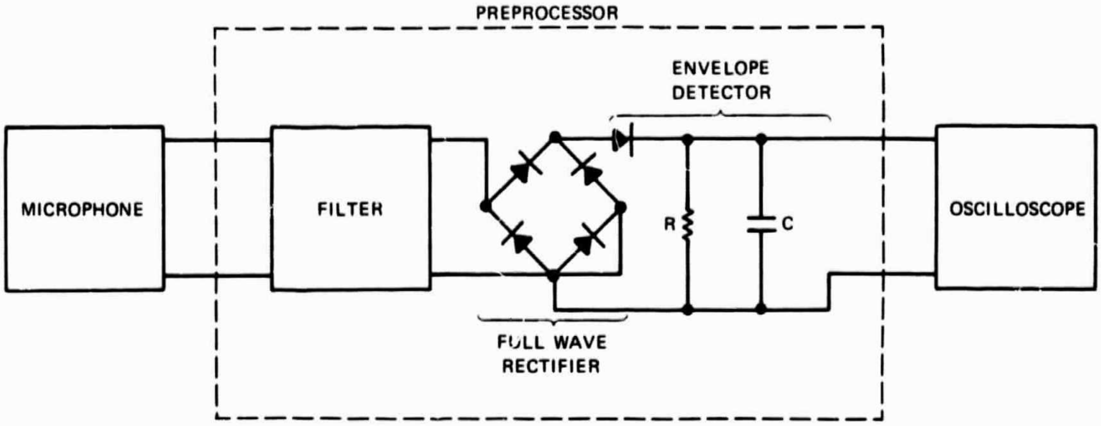


Figure 2

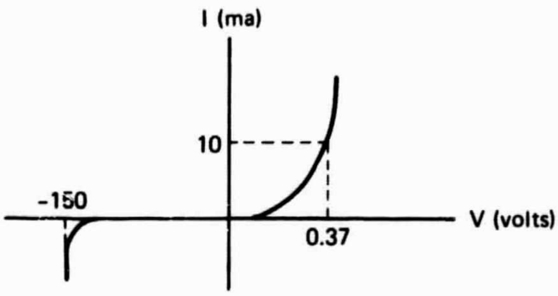
Idea #1



time constant of envelope detector = RC

The voltage across the capacitor may have an undesired ripple frequency which may be filtered out by a low-pass filter.

Diodes used for full wave rectifier



rectifier diodes and D1 : DR435 diodes

C = 20 ufd, electrolytic capacitor

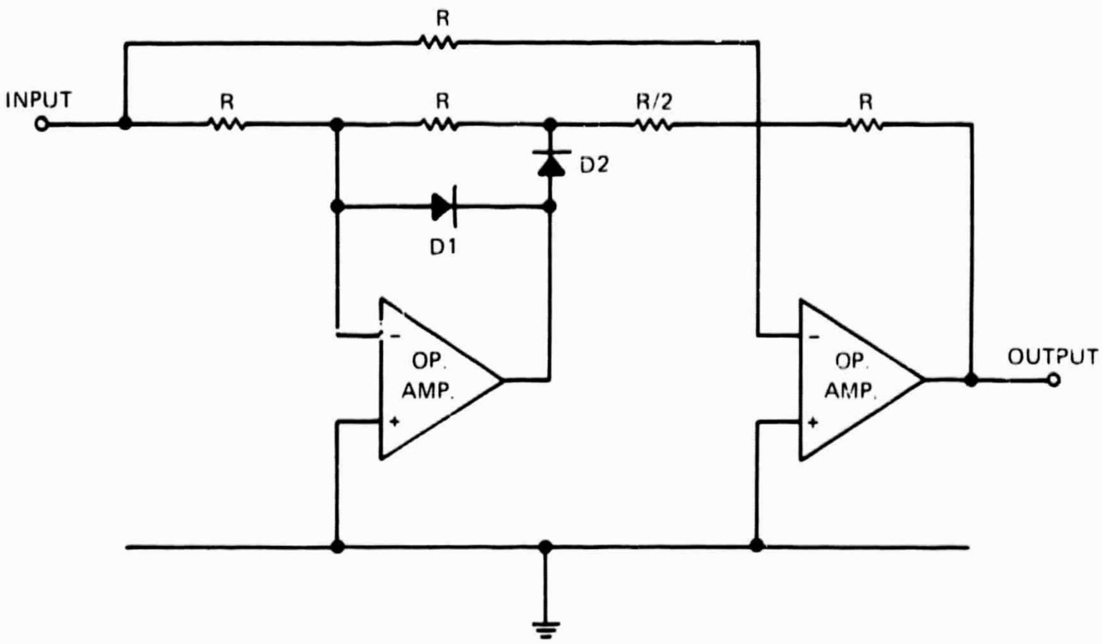
R = ranges from 0 to 5 M ohms

changing the value of R the time constant varies from 0 to 20 sec.

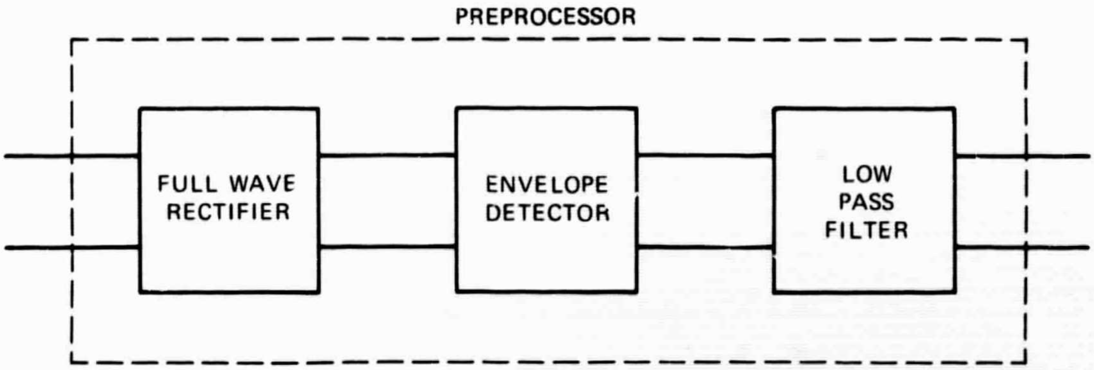
Problem with the four diode full wave rectifier:

It is desirable to have both sides of diode D2 above grounded. With both sides of this diode grounded there will not be full wave rectification making it necessary to find another full wave rectifier.

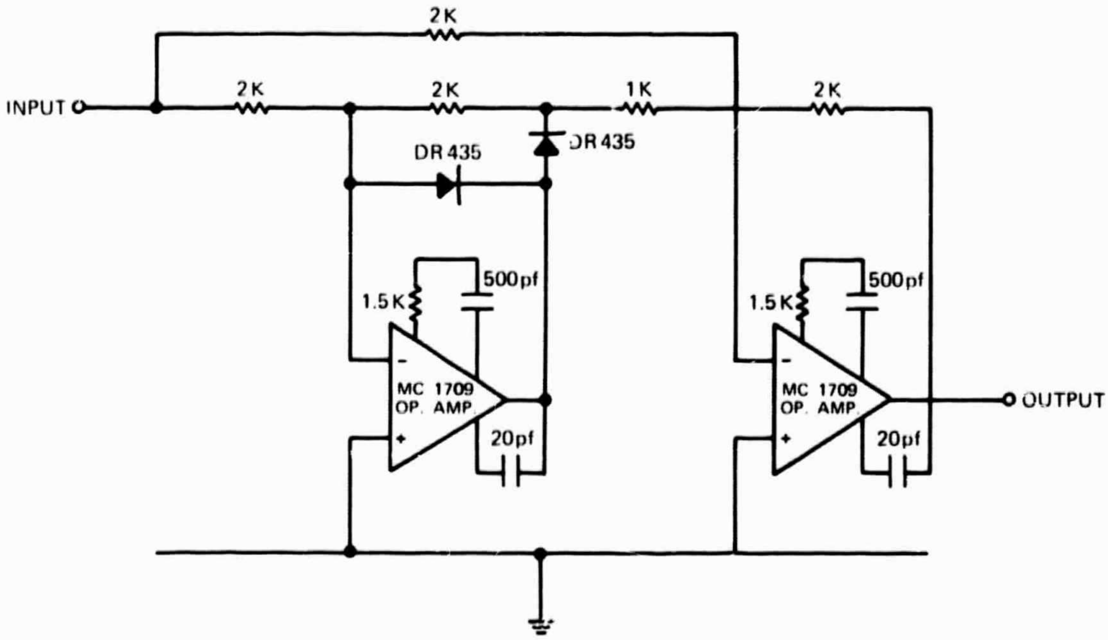
Full Wave Rectifier Found in Burr-Brown Application Notes



August 27, 1970



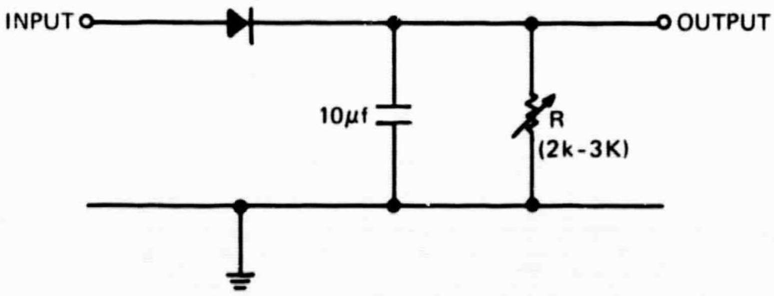
Full Wave Rectifier Used:



This full wave rectifier is good at all of the frequencies that are needed to rectify heart sounds. It has a low output impedance and there is no grounding problem which makes it a good match for the next stage of the preprocessor.

August 28, 1970

Envelope Detector:



This envelope detector simply smooths out the rectified signal. It does not filter out any part of the signal that is needed retaining the information from all the frequencies contained in a heart sound.

An RC time constant is chosen so that the necessary diagnostic information is retained while at the same time the waveform is smoothed. Time constant used = RC = 0.02 to 0.03 seconds.

On the positive cycle of the input signal, the capacitor C (10 uf) charges up to the peak voltage of the input signal. As the input signal falls below the peak value, the diode is cut off because the capacitor voltage (which is very nearly the peak voltage) is greater than the input signal voltage, thus causing the diode to open. The capacitor then discharges through the resistor R. During the next positive cycle, near the peak of the input signal, the input signal becomes greater than the capacitor voltage and the diode conducts. The capacitor again charges to the peak value of this new cycle. During the cutoff period the capacitor will discharge completely with no new input.

Low Pass Filter

"Frequency selective networks for use in the frequency range below 100 kHz have always been a problem. In this area of operation the inductors and capacitors required are large, both in value and physical size. Also, at these frequencies inductors and capacitors become quite lossy and the circuit Q's begin to suffer.

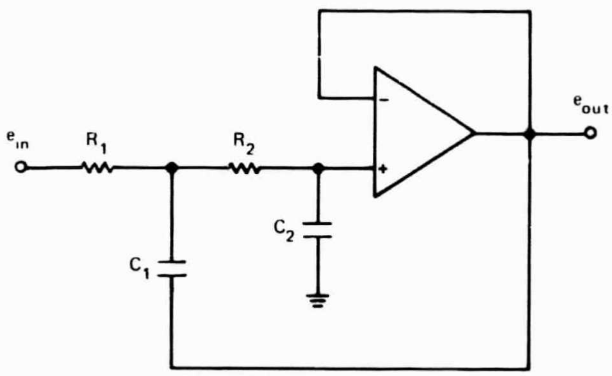
"The answer to this problem is to exchange the large inductor and capacitor for a large block of gain and use well known feedback principles to achieve selectivity with R-C active filters. Previously, to achieve a high degree of accuracy and circuit stability, a large number of active components was required in a fairly sophisticated circuit. Consequently, the design time and number of active components required made the use of active filters quite expensive.

"The solution to this problem came with the advent of integrated circuits which allowed transistors to be "less expensive" than resistors. Now, excellent gain blocks can be fabricated at fairly reasonable costs. And as technology improves, the performance will continue to improve and the costs will continue to decline, making the use of active filters very economical."

- - - - - from Motorola Application Notes

After rectifying the heart sounds and putting the rectified wave into the envelope detector, we put the output of the envelope detector into a Rockland filter to determine what cutoff frequency was needed for a smooth envelope. This filter is filtering the envelope, not the heart sounds. Desired cutoff frequency = $35 \ \frac{\text{cycles}}{\text{sec.}}$

Second Order Low Pass Active Filter



Configuration from:

OPAMP LABS
172 S. Alta Vista Blvd.
Los Angeles, Calif. 90003
(213) 934-3566
Application notes for
Model 4009 Medium Voltage
D.C. Operational Amplifier

$$\frac{e_o}{e_i} = \frac{1}{\left(\frac{S}{Wn}\right)^2 + {}^{d}\!\left(\frac{S}{Wn}\right) + 1}$$

where $$d = \frac{2 + R_1C_2}{1\ 2}$$

$$Wn^2 = \frac{1}{2\ 2} \qquad \begin{aligned} 1 &= R_1\, C_1 \\ 2 &= R_2\, C_2 \end{aligned}$$

want $$n = 35\ \frac{\text{cycles}}{\text{sec.}} \qquad \text{so } Wn = 2 \qquad = \frac{\text{rad.}}{\text{sec.}}$$

pick $$R_1 = 120\text{K and } C_1 = 0.05$$

$$1 = R_1C_1 = (1.2 \times 10^5)\ (0.05 \times 10^{-6}) = 6 \times 10^{-3} \text{ sec.}$$

$$Wn^2 = \frac{1}{1\ 2} = \frac{1}{(6 \times 10^{-3})}\ 2 = 4.84 \left(\frac{\text{rad.}}{\text{sec.}}\right)^2\ 2 \times 10^4$$

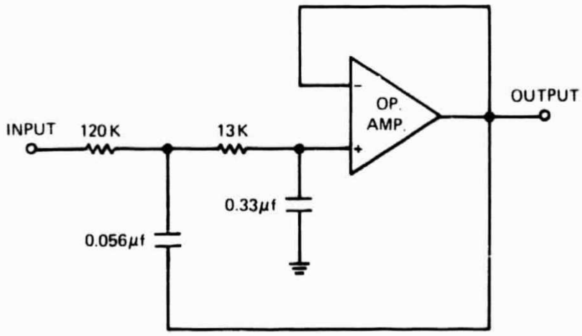
$$2 = \frac{1}{(6 \times 10^{-3})\ (4.84 \times 10^4)} = 3.4 \times 10^{-3} \text{ sec.}$$

pick $$R_2 = 13\text{K}$$

$$r_2 = R_2C_2$$

$$C_2 = \frac{2}{R_2} = \frac{3.4 \times 10^{-3}}{1.3 \times 10^4} = 0.26$$

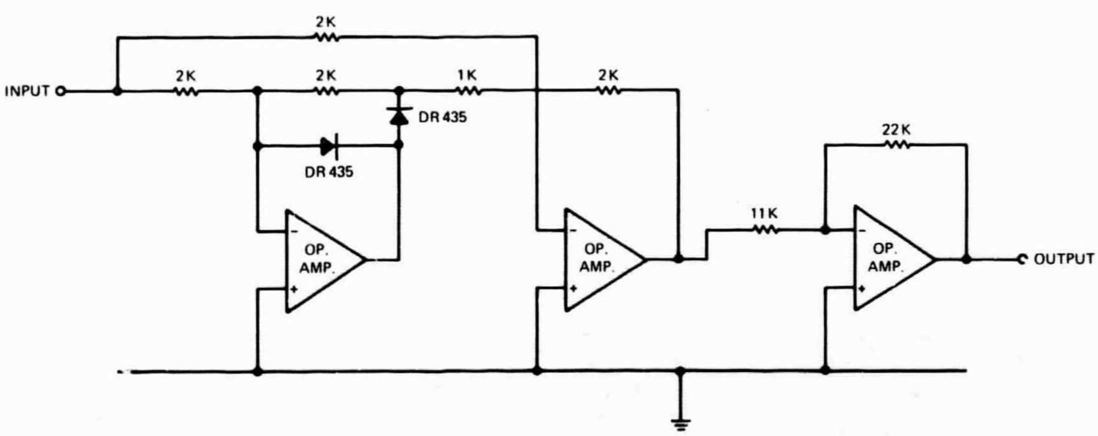
Low pass active filter used for preprocessor (from calculations on previous page)



Operational amplifiers use:
Model 4009 Medium
Voltage D.C. Operation
Amplifier from:
OPAMP LABS
172 S. Alta Vista Blvd.
Los Angeles, Calif. 90036
(213) 934-3566

Cutoff frequency determined in the lab = 33 $\frac{\text{cycles}}{\text{sec.}}$

Latest full-wave rectifier used:



Operational amplifiers used: Burr Brown

August 20, 1970

Note from Innocent Murmurs

Murmurs -

The intensity of the murmur is related to the velocity of blood flow and according to one theory intensity varies as the fourth power of velocity of flow.

Velocity is dependent on factors like volume of shunt, cardiac output, etc. So, if a small shunt and low velocity of blood exists the murmur will be of low intensity.

August 28, 1970

AM Approach for "Demodulation" of Heart Sounds

From the meetings I had with Mr. Mark Wilbur the following points became clear about this method and the way it had been used.

1. Assume heart sound is an amplitude modulated signal and try to demodulate it.

2. Pick a suitable carrier frequency to do the demodulation.

3. Pick a suitable filter (cutoff frequency and voll. off) for blocking redundant information.

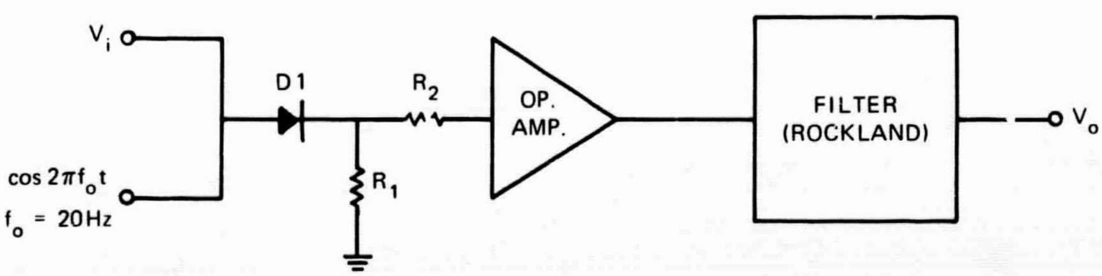
4. Devise a method to multiply the carrier into the AM signal.

The carrier frequency was picked at 20 Hz in order to demodulate the signal by sending the lowest end of the audible range of the spectrum. Any frequency lower than 20 Hz cannot be heard, therefore, we do not (it was assumed) need that part of the frequencies of the signal in the envelope.

The filter was picked as low pass filter with the cut off point at 17 Hz with a roll off of 24 db/octave. This choice was made by looking at the output of the filter and changing the cutoff frequency and the roll off until a reasonable envelope was achieved.

The device to multiply the signal with the carrier signal was chosen to be a diode.

The simplified circuit which was used to get the result was the following.



The assumption was that the diode in the above circuit could behave as a multiplier. Current in the forward direction of the diode $I_f = V + \frac{V^2}{2!} + \frac{V^3}{3!} \ldots$ using the exponential characteristic of a diode where V is the voltage across it.

$$I_f = (V_i + \cos 2\pi f_0 t) + \frac{(V_i + \cos 2\pi f_0 t)^2}{2!} + \frac{(V_i + \cos 2\pi f_0 t)^3}{3!} + \ldots$$

$$I_f \cong V_i + \cos 2\pi f_0 t + \frac{V_i^2}{2} + \frac{\cos^2 2\pi f_0 t}{2} + \cos 2\pi f_0 t\, V_i$$

$$I_f \cong V_i + \cos 2\pi f_0 t + \frac{V_i^2}{2} + \frac{1 + \cos 4\pi f_0 t}{4} + V_i \cos 2\pi f_0 t$$

This current goes through R_l and converts itself to a voltage value to go into the high impedance Op Amp to eventually go through the filter. The filter having a cut off frequency at 17 Hz blocks the $\cos(2\pi f_0 t)$ obviously, blocks $\frac{\cos(4\pi f_0 t)}{4}$ term blocks most part of $\frac{V_i^2}{2}$ (because $\frac{V_i^2}{2}$ is formed of higher frequencies because of the fact that most of V_i spectrum is above 20 Hz, so most of $\frac{V_i^2}{2}$ will be above 20 Hz.) Most of the v_i term will be blocked too, so we will end up with the V_i.

The results which were obtained from this method were OK because the output of the filter looked like the envelope of the incoping signal with approximation.

The high frequency murmurs were not being detected and the low intensity portion of the signal was not detected well. The important point was that the patient's murmurs were usually of higher frequencies and sometimes of low intensity. So, the diagnostic value of the envelope was almost very small, if any.

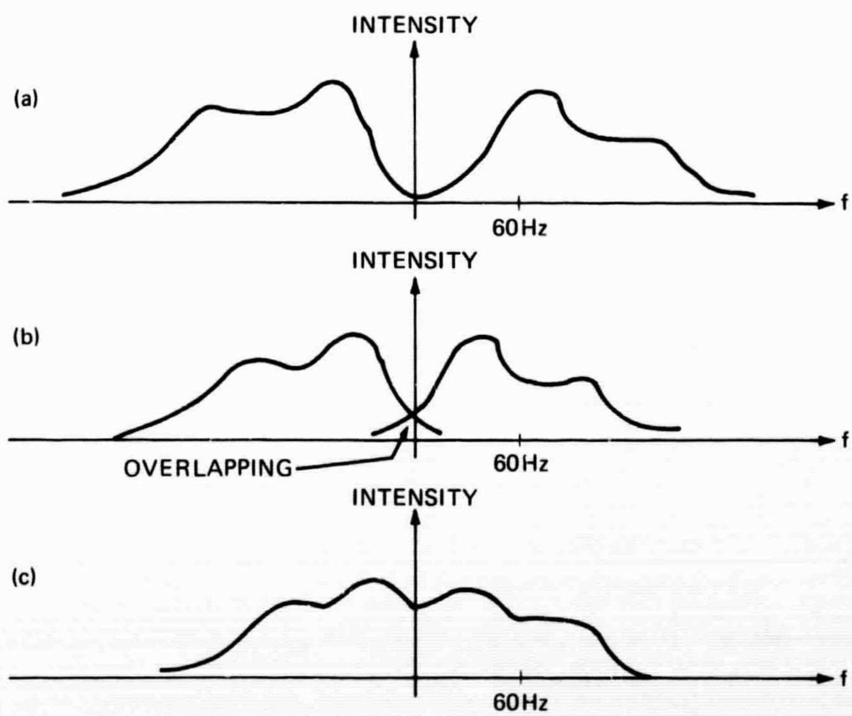
In order to get better results from this method we replaced the diode with a real multiplier. The multiplier was a Fast Quarter-Square Multiplier/Divider. It was a definite improvement over a diode. The diode was picked on the assumption that we could ignore the higher terms in the Taylor Series of the exponential curve of the diode. This is usually not a valid assumption with the kind of signal we have here. The voltage goes high and low and we know when V is not very small. The assumption of ignoring the higher terms is invalid. Furthermore, in order to get better results we decided to improve the choice of carrier frequency and also filtering.

The Rockland Filter was used as a frequency analyzer to determine a spectrum knowledge (intensity vs time) of the heart sound. The bandwidth of 20 Hz was picked and the intensity of the sound was measured in this bandwidth as it was swept across all frequencies involved (0-1000 Hz). Many curves were obtained for different heart sounds. The conclusion was most of the spectrums were very different from each other, but the fact was that most of them had high intensity components (usually belonging to first and second sounds) between 40 Hz and

100 Hz. This knowledge determined the choice of carrier frequency for us. It was picked to be 40 Hz. This would shift the "important" part of the spectrum down to D.C. and around it. The filter cutoff was at 25 Hz with a roll off of 24 db/octave. The result was a signal similar to the envelope of the sound, but again very approximate and poor with respect to higher frequency murmurs.

One important mistake with the previous circuit is that because of the diode half of the sound signal is completely ignored. In any case, the second attempt with the AM approach failed too. This was because the spectrum of the sounds we achieved showed that no one frequency can be picked to demodulate the signal with, for the simple reasons that the complex signal received from the heart is not an amplitude modulated signal and it is not even close to something of that kind. The spectrums were very much distributed and there were no definite peaks to suggest choice of several carrier frequencies for "demodulation." This idea to pick different parts of the spectrum, shift them to DC and low frequencies and then add them to each other, fails to be a good one for several reasons. As mentioned above there is difficulty with choice of carrier frequencies. Then there is the problem of the negative frequency components of the Fourier Transform of the signal which gets shifted to DC and low frequencies each time.

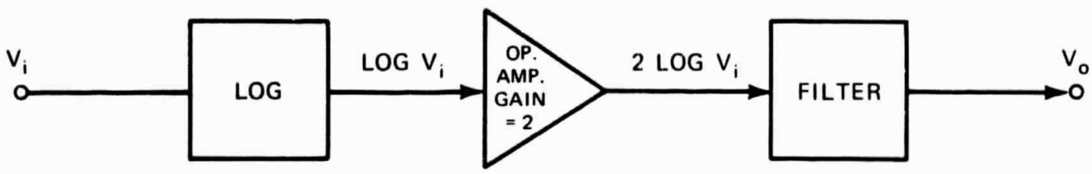
We multiply the signal with cosine of some frequency of time.



In (a) we have spectrum of a typical heart sound which is shifted down in (b) and the resultant spectrum is shown in (c). The overlapping of the negative frequency components introduces distortion of the results. This error could be very much amplified when several carrier frequencies are used for demodulating adding and getting the result. This important point of the addition of errors, the fact that the spectrum does not have well-defined peaks and it is distributed fairly smoothly makes it unwise to use the method of demodulating and adding the signals to get a good envelope.

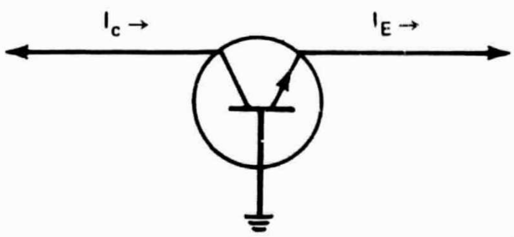
The Log Approach for Detection of Envelope of Heart Sounds

Ear is sensitive to log of intensity of the sounds, therefore, in order to get an envelope that looks like what the sound sounds like, we can initiate this biological process by electronics.



V_i is the rectified (full wave) of the heart sound signal. The Op Amp puts out 2 log V_i which is log V_i^2 which is log of intensity (proportional to it) and the filter smooths the output signal. A time constant or delay could be introduced at the output to simulate the refractory periods of audio sensations of the ear.

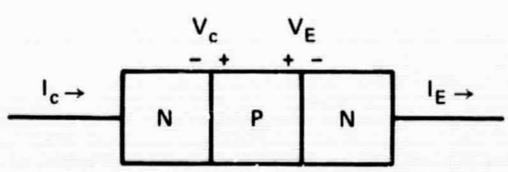
The circuit to get the log of the full wave rectified signal is based upon the property of exponential character of current voltage relationship in a transistor.



(a)

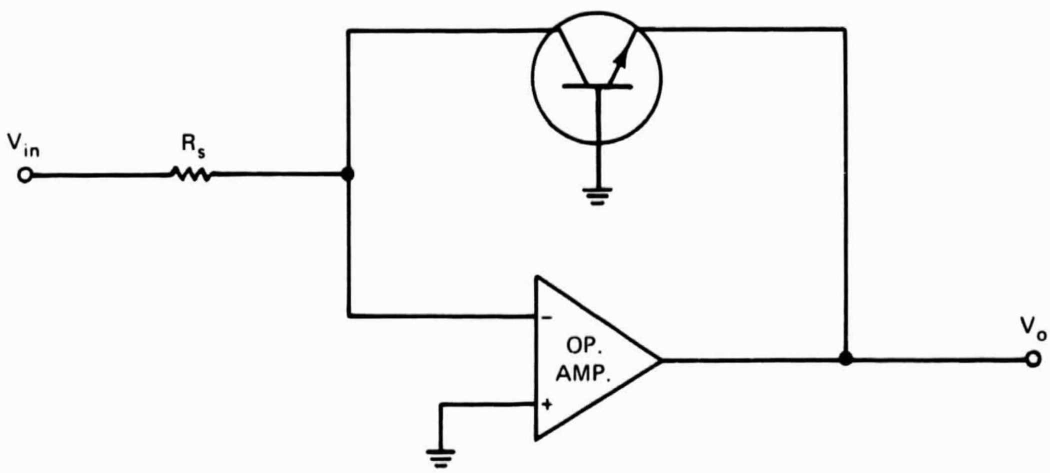
$$I_c = a_n I_{es} \left[e^{\zeta V_E/KT}\right]$$

I_{es} is the reverse saturation current of the emitter.



The basic circuit used is shown in Figure (b).

(b)



$$I_c = a_n I_{es} \left[e^{\zeta V_{E}/KT}\right] \qquad \text{(a)}$$

$$I_c = \frac{E_{in}}{R_s} \qquad \text{(b)}$$

Combining equations (a) and (b) we get

$$\frac{E_{in}}{R_s} = a n I_{es} \left[e^{\zeta V_{o}/KT}\right] \qquad V_o = \frac{KT}{\zeta} \ 1n \ \frac{E_{in}}{R_s a_n I_{es}}$$

Converting from $\log_e$ to $\log_{10}$ gives:

$$V_o = 2.3 \ \frac{KT}{\zeta} \ \log_e \frac{E_{in}}{R_s \ a_n \ I_{es}}$$

$$V_o = 2.3 \ \frac{KT}{\zeta} \ \log_{10} E_{in} + 2.3 \ \frac{KT}{\zeta} \ \log_{10} \frac{1}{R_s \ a_n \ I_{es}}$$

$$\text{at } T = 27°C \ \frac{KT}{\zeta} = 0.026V \qquad V_o = 0.06 \log_{10} E_{in} + K$$

Empirically it was found that V_o and E_i have the following relationship:

$$V_o = 0.062 \log_{10} (E_{in}) + 0.450$$

V_o and E_i are in volts.

We tried this method, but did not succeed to have a meaningful output. This was probably because we should have had extra circuits for frequency compensation of the operational amplifier.

The correct circuit and information is given in Application Note AN-261 of Motorola Semiconductor Products, Inc.

Task III
PHONOCARDIOGRAPHIC PREPROCESSOR

COMMENTS

Methodology

No formal outline was set up for the project. However, the material is well ordered in narrative form in the laboratory notebook. Previous work was reviewed and discussion was frequent with the Department of Clinical Engineering personnel. The various methods of intensity detection, such as square law detection by diodes, temperature devices and frequency shifting, were reviewed and discussed in the laboratory notebook.

Electronic circuits were breadboarded in NASA laboratories and parts were supplied by NASA. Various parameters of the circuit were measured and the data evaluated.

Results

A breadboard circuit consisting of a full wave rectifier with simple resistance capacity filtering was demonstrated. This circuit did produce an envelope of the phonocardiogram. The envelope, however, seems to have excessive time constant and was lacking in defining many of the characteristics of the phonocardiogram. Actual phonocardiographic recording on 1/4" magnetic tape was used to develop the circuit. It was interesting to note on this project, that the solution tended to be a sophisticated engineering approach, while the users (Clinical Engineering) saw a simplified empiric solution. This is a common theme in the area of medical instrumentation. This does not mean that instrumentation should not be technically sound, but rather that simplified solutions of less accuracy are many times acceptable.

Conclusion

The conclusion of the project, although not specifically stated, was that full wave rectification with an appropriate time constant circuit is feasible for the detection of phonocardiograms.

Future Application/Expansion

Work has continued on this project at the Department of Clinical Engineering. A circuit called a "box-car detector" was added to the original breadboard circuit

of the full wave rectifier. This seems to have less of a "time-constant" problem. The signal still needs the appropriate filtering. The filter characteristics will be developed using a computer program on sampled data input to simulate the response with the results reviewed by physicians and engineers until an acceptable envelope is presented for the appropriate disease categories.